AMENDMENTS TO THE CLAIMS

Claims 1, 2, 10, 12 and 22 are currently being amended, and claim 5, 6, 15 and 16 are being canceled. All pending claims are reproduced below.

1. (Currently Amended) A method comprising:

(a) storing a plurality of independent sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions;

(b) storing a value in a filter selection register;

(c) selecting a single one of the independent sets of filter coefficients based on the value stored in the filter selection register;

(d) receiving an audio input signal including a plurality of samples;

(e) estimating a sample rate of [[an]] the audio input signal;

(f) interpolating the single one selected set of filter coefficients, in dependence on the estimated sample rate of the audio input signal, to thereby produce interpolated polyphase filter coefficients; and

(g) convolving the produced interpolated polyphase filter coefficients with the samples of the audio input signal to produce a filtered audio output signal that differs from the audio input signal regardless of which single one of the sets of filter coefficients is selected;

wherein said selecting the single one of the independent sets of filter coefficients at step (c) is performed prior to receiving the audio input signal at step (d), independent of the audio input signal received at step (d), and independent of the filtered audio output signal produced at step (g); and

wherein the same single one of the sets of filter coefficients selected at step (c) is used at steps (f) and (g) to produce the filtered audio output signal produced at step (g).

2. (Currently Amended) The method of claim 1, ~~wherein the input signal comprises an audio signal,~~ wherein the audio input signal is convolved with the interpolated filter coefficients in a sample rate converter of a digital pulse width modulation (PWM) audio amplifier.

3.-6. (Canceled)

7. (Original) The method of claim 1, wherein the plurality of sets of filter coefficients are stored in a single memory.

8. (Previously Presented) The method of claim 1, wherein the single one selected set of filter coefficients are interpolated according to a cubic spline algorithm.

9. (Original) The method of claim 1, wherein each of the plurality of sets of filter coefficients comprise polyphase filter coefficients.

10. (Currently Amended) A system comprising:

a coefficient interpolator;

a filter selection register;

a memory coupled to the coefficient interpolator; and

a sample rate estimator configured to estimate a sample rate of an audio input signal;

wherein the memory is configured to store multiple independent sets of filter coefficients, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions; and

wherein the coefficient interpolator is configured to interpolate a selected single one of the independent sets of filter coefficients, in dependence on the estimated sample rate of the audio input signal, to thereby produce interpolated polyphase filter coefficients; and

wherein the selected single of one of the independent sets of filter coefficients is selected based on contents of the filter selection register, independent of the audio input signal received, and independent of the filtered audio output signal produced; and

wherein the same single one of the sets of filter coefficients selected is used to produce the filtered audio output signal.

11. (Previously Presented) The system of claim 10, further comprising a convolution engine coupled to the coefficient interpolator and configured to convolve the input signal with the produced interpolated polyphase coefficients corresponding to the selected single one of the sets of filter coefficients to produce an output signal that differs from the input signal regardless of which one of the sets of filter coefficients is selected.

12. (Currently Amended) The system of claim 11, wherein:

~~the input signal comprises an audio input signal; and~~

the convolution engine is implemented in a sample rate converter of a pulse width modulation (PWM) amplifier.

13.-18. (Canceled)

19. (Original) The system of claim 10, wherein the memory comprises a single memory module configured to store the multiple sets of filter coefficients.

20. (Previously Presented) The system of claim 19, wherein each of the multiple independent sets of filter coefficients comprise polyphase filter coefficients.

21. (Original) The system of claim 10, wherein the coefficient interpolator is configured to interpolate the selected set of filter coefficients according to a cubic spline algorithm.

22. (Currently Amended) A method comprising:

storing a plurality of independent sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions;

storing a value in a filter selection register;

selecting a single one of the sets of filter coefficients based on the value stored in the filter selection register;

estimating a sample rate of an input signal;

receiving an audio data signal and frame sync signals associated with the audio data signal;

estimating, based on the frame sync signals, a sample rate of audio data signal;

interpolating the single one selected set of filter coefficients, in dependence on the estimated sample rate of the input signal, to thereby produce interpolated polyphase filter coefficients; and

convolving the produced interpolated polyphase filter coefficients with the received audio data signal to produce a filtered audio data signal that differs from the received audio data signal regardless of which single one of the sets of filter coefficients is selected;

wherein said selecting the single one of the independent sets of filter coefficients is performed prior to receiving the audio input audio signal, independent of the audio input signal received, and independent of the filtered audio output signal produced; and

wherein the same single one of the sets of filter coefficients selected is used to produce the filtered audio output signal.

23. (Previously Presented) The method of claim 22, further comprising performing the method in a sample rate converter of a digital PWM amplifier.

24. (Previously Presented) The method of claim 1, wherein the plurality of sets of filter coefficients are stored in the memory prior to receiving the input signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in the output signal.

25. (Previously Presented) The system of claim 10, wherein the memory is configured to store the multiple sets of filter coefficients prior to receiving an input signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in an output signal produced by convolving the input signal with interpolated coefficients based on the corresponding set of filter coefficients.

26. (Previously Presented) The method of claim 22, wherein the plurality of sets of filter coefficients are stored in the memory prior to receiving the audio data signal, and wherein the filter function defined by each set of filter coefficients corrects distortion in the produced filtered audio signal.

27. (Previously Presented) The method of claim 1, wherein the output signal, resulting from the convolving step, is dependent on which single one of the independent sets of filter coefficients is selected, such that for the same input signal a different output signal would be produced if a different one of the independent sets of filter coefficients were selected.

28. (Previously Presented) The system of claim 11, wherein the output signal, produced by the convolution engine, is dependent on which single one of the independent sets of filter coefficients is selected, such that for the same input signal a different output

signal would be produced if a different one of the independent sets of filter coefficients were selected.

29. (Previously Presented) The method of claim 22, wherein the filtered audio data signal, resulting from the convolving step, is dependent on which one of the independent sets of filter coefficients is the single one selected, such that for the same received audio data signal a different filtered audio data signal would be produced if a different one of the independent sets of filter coefficients were the single one selected.